Digital Communication

UNIT I
PULSE MODULATION

1. Define Dirac comb or ideal sampling function. What is its Fourier Transform?

   Dirac comb is nothing but a periodic impulse train in which the impulses are spaced by a time interval of Ts seconds. The equation for the function is given by

$$\delta_{Ts}(t) = \sum_{n=-\infty}^{\infty} \delta(t - n\,Ts)$$

The Fourier Transform of $\delta_{Ts}(t)$ is given by

$$F[\delta_{Ts}(t)] = f_s \sum_{m=-\infty}^{\infty} \delta(f - m f_s)$$

2. Give the interpolation formula for the reconstruction of the original signal g(t) from the sequence of sample values {g(n/2W)}.

$$g(t) = \sum_{n=-\infty}^{\infty} g(n/2W)\ \mathrm{sinc}\,(2Wt - n)$$

where 2W is the bandwidth

n is the number of samples.

3. State sampling theorem.

   - If a finite –energy signal g(t) contains no frequencies higher than W hertz ,it is completely determined by specifying its co=ordinates at a sequence of points spaced 1/2W seconds apart.
   - If a finite energy signal g(t) contains no frequencies higher than W hertz, it may be completely recovered from its co=ordinates at a sequence of points spaced 1/2W seconds apart.

4. Define quadrature sampling.

   Quadrature sampling is used for uniform sampling of band pass signals

   Consider $g(t) = g_I(t)\cos(2\Pi fct) - g_Q(t)\sin(2\Pi fct)$.

The in-phase component $g_I(t)$ and the quadrature component $g_Q(t)$ may be obtained by multiplying the bandpass signal g(t) by cos(2Π fct) and sin(2Πfct) respectively and then suppressing the sum-frequency components by means of appropriate low pass filter. Under the assumption that fc>W,we find that $g_I(t)$&$g_Q(t)$ are both low-pass signals limited to -W<f<W. Accordingly each component may be sampled at the rate of 2W samples per second. This type of sampling is called quadrature sampling.

5. What is aliasing?

The phenomenon of a high-frequency in the spectrum of the original signal g(t) seemingly taking on the identity of a lower frequency in the spectrum of the sampled signal g(t) is called aliasing or foldover.

6. Give the expression for aliasing error and the bound for aliasing error.

$$E = \left| \sum_{n=-\infty}^{\infty} [1-\exp(-j2\Pi m f_s t)] \int_{(m-1/2f_s)}^{(m+1/2f_s)} G(f) \exp(j2\Pi f t)\, df \right|$$

where E is the aliasing error.

|G(f)| is the amplitude spectrum of the signal g(t).

$$E < 2 \int_{|f| > f_s/2} |G(f)|\, df.$$

Where E is the bound for aliasing error.

7. What is meant by PCM?

Pulse code modulation (PCM) is a method of signal coding in which the message signal is sampled, the amplitude of each sample is rounded off to the nearest one of a finite set of discrete levels and encoded so that both time and amplitude are represented in discrete form.. This allows the message to be transmitted by means of a digital waveform.

8. Define quantizing process.

   The conversion of analog sample of the signal into digital form is called quantizing process.

9. What are the two fold effects of quantizing process.
   1. The peak-to-peak range of input sample values subdivided into a finite set of decision levels or decision thresholds
   2. The output is assigned a discrete value selected from a finite set of representation levels are reconstruction values that are aligned with the treads of the staircase.

10. What is meant by idle channel noise?

    Idle channel noise is the coding noise measured at the receiver output with zero transmitter input.

11. What is meant by prediction error?

    The difference between the actual sample of the process at the time of interest and the predictor output is called a prediction error.

12. Define delta modulation.

    Delta modulation is the one-bit version of differential pulse code modulation.

13. Define adaptive delta modulation.

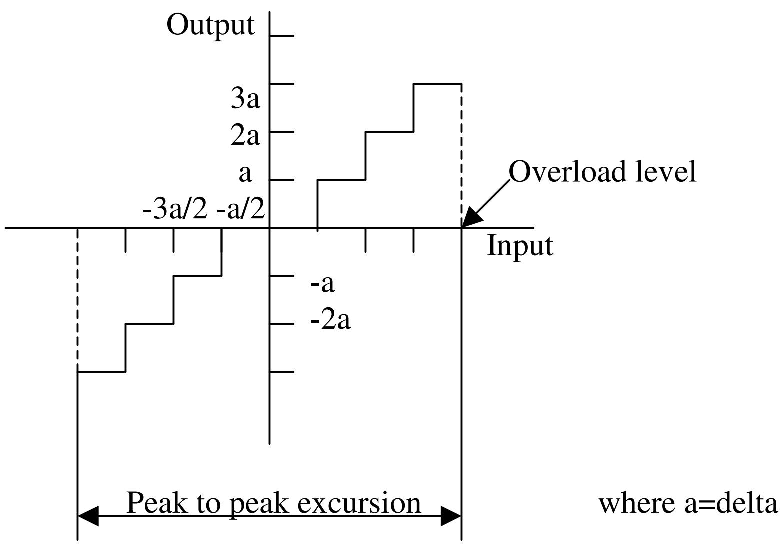
    The performance of a delta modulator can be improved significantly by making the step size of the modulator assume a time- varying form. In particular, during a steep segment of the input signal the step size is increased. Conversely, when the input signal is varying slowly, the step is reduced , In this way, the step size is adapting to the level of the signal. The resulting method is called adaptive delta modulation (ADM).

14. Name the types of uniform quantizer?

1. Mid tread type quantizer.
2. Mid riser type quantizer.

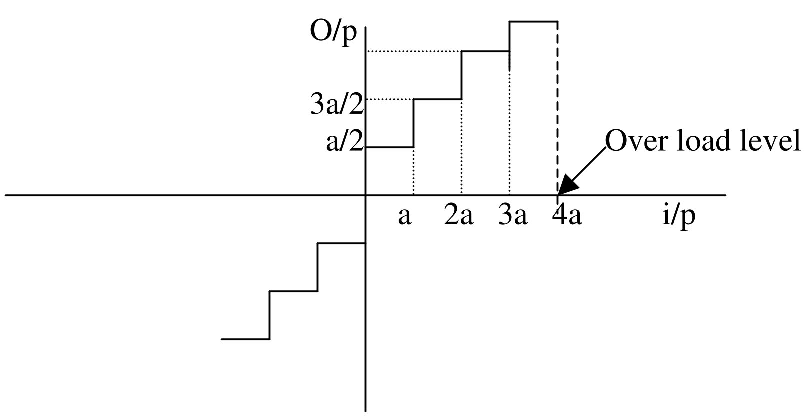
15. Define mid tread quantizer?

Origin of the signal lies in the middle of a tread of the staircase.



16. Define mid-riser quantizer?

Origin of the signal lies in the middle of a riser of the staircase



17. Define quantization error?

Quantization error is the difference between the output and input values of quantizer.

18. What you mean by non-uniform quantization?

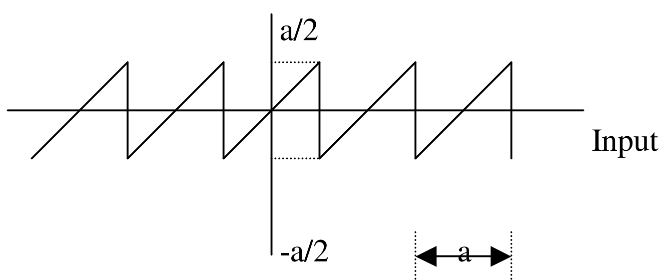
Step size is not uniform. Non-uniform quantizer is characterized by a step size that increases as the separation from the origin of the transfer characteristics is increased. Non-uniform quantization is otherwise called as *robust quantization*

.

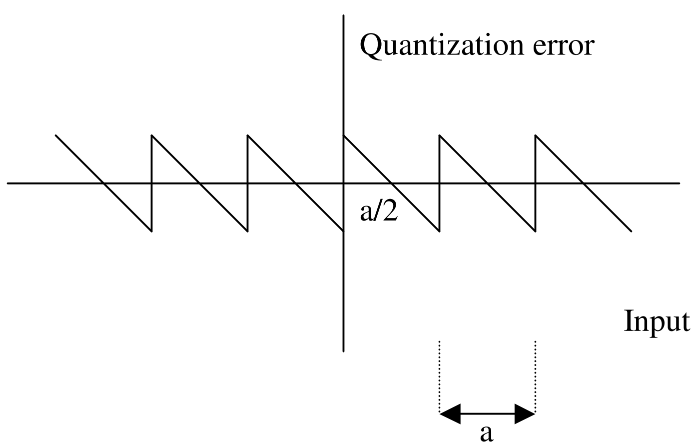
19. Draw the quantization error for the mid tread and mid-rise type of quantizer?

For mid tread type:

Quantization error



For mid riser type:



20. What is the disadvantage of uniform quantization over the non-uniform quantization?

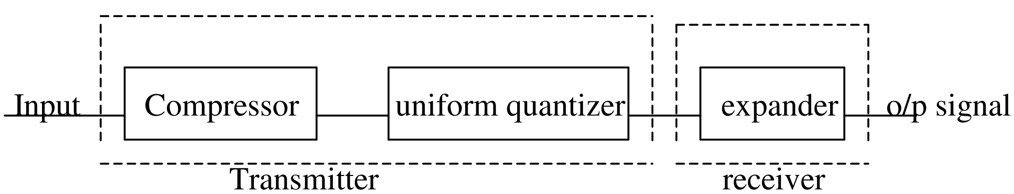
SNR decreases with decrease in input power level at the uniform quantizer but non-uniform quantization maintains a constant SNR for wide range of input power levels. This type of quantization is called as robust quantization.

21. What do you mean by companding? Define compander.

The signal is compressed at the transmitter and expanded at the receiver. This is called as *companding*. The combination of a compressor and expander is called a *compander.*

22. Draw the block diagram of compander? Mention the types of companding?

Block diagram:



Types of companding:

1. μ law companding
2. A law companding

23. What is PAM?

PAM is the pulse amplitude modulation. In pulse amplitude modulation, the amplitude of a carrier consisting of a periodic train of rectangular pulses is varied in proportion to sample values of a message signal.

24. What is the need for speech coding at low bit rates?

The use of PCM at the standard rate of 64 Kbps demands a high channel bandwidth for its transmission ,so for certain applications, bandwidth is at premium, in which case there is a definite need for speech coding at low bit rates, while maintaining acceptable fidelity or quality of reproduction.

25. Define ADPCM.

It means adaptive differential pulse code modulation, a combination of adaptive quantization and adaptive prediction. Adaptive quantization refers to a quantizer that operates with a time varying step size. The autocorrelation function and power spectral density of speech signals are time varying functions of the respective variables. Predictors for such input should be time varying. So adaptive predictors are used.

UNIT II

BASEBAND TRANSMISSION

26. What is meant by forward and backward estimation?

AQF: Adaptive quantization with forward estimation. Unquantized samples of the input signal are used to derive the forward estimates.

AQB: Adaptive quantization with backward estimation. Samples of the quantizer output are used to derive the backward estimates.

APF: Adaptive prediction with forward estimation, in which unquantized samples of the input signal are used to derive the forward estimates of the predictor coefficients.

APB: Adaptive prediction with backward estimation, in which Samples of the quantizer output and the prediction error are used to derive estimates of the predictor coefficients.

27. What are the limitations of forward estimation with backward estimation?

- Side information
- Buffering
- Delay

28. How are the predictor coefficients determined?

For the adaptation of the predictor coefficients the least mean square (LMS) algorithm is used.

29. Define adaptive subband coding?

It is a frequency domain coder, in which the speech signal is divided in to number of subbands and each one is coded separately. It uses non masking phenomenon in perception for a better speech quality. The noise shaping is done by the adaptive bit assignment.

30. What are formant frequencies?

In the context of speech production the formant frequencies are the resonant frequencies of the vocal tract tube. The formants depend on the shape and dimensions of the vocal tract.

31. What is the bit rate in ASBC?

Nfs= (MN) (fs/M)

Nfs->bit rate

M->number of subbands of equal bandwidths

N->average number of bits

fs/M->sampling rate for each subband

32. Define Adaptive filter?

It is a nonlinear estimator that provides an estimate of some desired response without requiring knowledge of correlation functions, where the filter coefficients are data dependent. A popular filtering algorithm is the LMS algorithm.

33. Define data Signalling Rate.

Data signalling rate is defined as the rate measured in terms bits per second(b/s) at which data are transmitted.

Data signaling rate Rb=I/Tb

Where Tb=bit duration.

34. Define modulation rate.

It is defined as the rate at which signal level is changed depending On the nature of the format used to represent the digital data.It is measured in Bauds or symbols per second.

35. State NRZ unipolar format

In this format binary 0 is represent by no pulse and binary 1 is Represented by the positive pulse.

36. State NRZ polar format.

Binary 1 is represented by a positive pulse and binary 0 is represented by a Negative pulse.

37. State NRZ bipolar format.

Binary 0 is reporesented by no pulse and binary one is represented by the alternative positive and negative pulse.

38. State Manchester format.

Binary 0 → The first half bit duration negative pulse and the second half Bit duration positive pulse.

Binary 1→ first half bit duration positive pulse and the second half Bit duration negative pulse.

39. What is an eye pattern?

Eye Pattern is used to study the effect of intersymbol interference.

40. What is the width of the eye?

It defines the time interval over which the received waveform can be sampled without error from intersymbol interference.

41. What is sensitivity of an eye?

The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied.

42. What is margin over noise?

The height of the eye opening at a specified sampling time defines the margin over noise.

43. What is Inter symbol interference?

The transmitted signal will undergo dispersion and gets broadened during its transmission through the channel. So they happen to collide or overlap with the adjacent symbols in the transmission. This overlapping is called Inter Symbol Interference.

44. How eye pattern is obtained?

The eye pattern is obtained by applying the received wave to the vertical deflection plates of an oscilloscope and to apply a saw tooth wave at the transmitted symbol rate to the horizontal deflection plate.

UNIT III
PASSBAND TRANSMISSION

45. Properties of matched filter.

The signal to noise ratio of the matched filter depends only upon the the ratio of the signal energy to the psd of white noise at the filter input

1) The output signal of a matched filter is proportional to a shifted version of the auto_correlation function of the input signal to which the filter is matched.

46. Why do we go for Gram-Schmidt Orthogonalization procedure?

Consider a message signal m. The task of transforming an incoming message mi=1,2,…..M, into a modulated wave si(t) may be divided into separate discrete time & continuous time operations. The justification for this separation lies in the Gram-Schmidt orthogonalization procedure which permits the representation of any set of M energy signals, {si(t)}, as linear combinations of N orthonormal basis functions, where $N \leq M$.

47. What is matched filter receiver ?

A filter whose impulse response is a time reversed & delayed version of some signal ∅ j (t) then it is said to be matched to ∅j (t) correspondingly, the optimum receiver based on the detector is referred to as the matched filter receiver.

48. What is maximum likelihood detector.

Maximum likelihood detector computes the metric for each transmitted message compares them and then decides in favor of maximum. The device for implementing the decision rule

i.e; set ^m = m_i if

In [f_x(x/mk)] is maximum for k=i is called maximum –likelihood detector and the decision rule is called maximum likelihood.

49. Define antipodal signals.

A pair of sinusoidal signals that differ only in a phase shift of 180 degrees are referred to as antipodal signals.

50. Explain how QPSK differs from PSK in term of transmission bandwidth and bit information it carries?

For a given bit rate $1/T_b$, a QPSK wave requires half the transmission bandwidth of the corresponding binary PSK wave. Equivalently for a given transmission bandwidth, a QPSK wave carries twice as many bits of information as the corresponding binary PSK wave

51. Give the equation for average probability of symbol error for coherent binary PSK.

Average probability of signal error,

$P_e = 1 / 2 \ \mathrm{erfc}\sqrt{E_b / N_o}$

52. Give the signal space characterization of QPSK.

i/p dibit $0 \le t \le T$	Phase of QPSK signal (radians)	Coordinates message points	
		S_{i1}	S_{i2}
10	∏/ 4	$\sqrt{E/2}$	$-\sqrt{E/2}$
00	3∏/4	$-\sqrt{E/2}$	$-\sqrt{E/2}$
01	5∏/4	$-\sqrt{E/2}$	$\sqrt{E/2}$
11	7∏/4	$\sqrt{E/2}$	$\sqrt{E/2}$

53. Define QPSK.

QPSK is Quadriphase –shift keying. In QPSK the phase of the carrier takes on one of the four equally spaced values Such as ∏/4 , 3∏/4, 5∏/4 and 7∏/4.

54. Define Dibit.

A unique pair of bits is called a dibit. Gray encoded set of dibits 10, 00, 01 & 11

55. Give the transmitted signal of Non-coherent binary FSK.

$$Si(t) = \begin{cases} \sqrt{2Eb/Tb}\ \mathrm{Cos}(2\prod f\, i\, t), & 0 \le t \le Tb \\ O, & \text{elsewhere} \end{cases}$$

fi = nc+ i/ Tb

56. Give the two basic operation of DPSK transmitter.

1. differential encoding of the input binary wave
2. Phase –shift keying hence, the name differential phase shift keying

57. Define deviation ratio in MSK .

The parameter h is defined by

$$h= Tb(f1-f2)$$

h is deviation ratio , measured with respect to bit rate 1/Tb.

58. Define MSK signal in interval $0\le t \le Tb$.

$$S(t) = \sqrt{2Eb/Tb}\ \mathrm{Cos}\ [\ 2\prod f\ 1t + \theta(0)]\ \text{ for symbol 1}$$
$$\sqrt{2Eb/Tb}\ \mathrm{Cos}\ [\ 2\prod f\ 2\ t + \theta(0)]\ \text{ for symbol 0}$$

59. What is nominal carrier frequency in MSK ?

Nominal carrier frequency is the arithmetic mean of the two frequencies f1 and f2 and it is given as

$f_c = \ ½ (f_1 + f_2)$

Where f_1 is the frequency for symbol –1

f_2 is the frequency for symbol – 0

60. What are the three broad types of synchronization ?

1. Carrier synchronization
2. Symbol & Bit synchronization
3. Frame synchronization.

61. What is carrier synchronization ?

The carrier synchronization is required in coherent detection methods to generate a coherent reference at the receiver. In this method the data bearing signal is modulated on the carrier in such a way that the power spectrum of the modulated carrier signal contains a discrete component at the carrier frequency.

62. What are the two methods for carrier synchronization.

1. Carrier synchronization using M^{th} Power loop
2. Costas loop for carrier synchronization

63. What is called symbol or bit synchronization ?

In a matched filter or correlation receiver, the incoming signal is sampled at the end of one bit or symbol duration. Therefore the receiver has to know the instants of time at which a symbol or bit is transmitted. That is the instants at which a particular bit or symbol status and when it is ended. The estimation of these times of bit or symbol is called symbol or bit synchronization.

64. What are the two methods of bit and symbol synchronization.

1) Closed loop bit synchronization
2) Early late gate synchronizer

65. What are the disadvantages of closed loop bit synchronization.

1) If there is a long string of 1's and o's then y(t) has no zero crossings and synchronization may be lost.
2) If zero crossing of y(t) are not placed at integer multiples of Tb, the synchronization suffers from timing Jitter.

66. What is called frame synchronization ?

Depending on bits used for encoding, the word length is defined. Thus each word container some fixed number of bits. The receiver has to know when a particular frame status and when its individual message bits status. This type of synchronization is called frame synchronization.

67. Why synchronization is required ?

The signals from various sources are transmitted on the single channel by multiplexing. This requires synchronization between transmitter and receiver. Special synchronization bits are added in the transmitted signal for the purpose. Synchronization is also required for detectors to recover the digital data properly from the modulated signal.

UNIT IV

ERROR CONTROL CODING

68. What is linear code ?

A code is linear if the sum of any two code vectors produces another code vector.

69. What is code rate ?

Code rate is the ratio of message bits (k) and the encoder output bits (n). It is defined by r (i.e) r= k/N

70. Define code efficiency.

It is the ratio of message bits in a block to the transmitted bits for that block by the encoder i.e

$$\text{Code efficiency} = \frac{\text{Message bits in a block}}{\text{Transmitted bits for the block}}$$

71. What is hamming distance?

The hamming distance between two code vectors is equal to the number of elements in which they differ. For example let the two code vectors be

X=(101) and Y= (110)

These two code vectors differ in second and third bits.

Therefore the hamming distance between x and Y is two.

72. What is meant by systematic & non-systematic code?

In a systematic block code, message bit appear first and then check bits. In the non-systematic code, message and check bits cannot be identified in the code vector.

73. How syndrome is calculated in Hamming codes and cyclic codes ?

In Hamming codes the syndrome is calculated as ,

$$S = YH^{T}$$

Here Y is the received and H^T is the transpose of parity check matrix.

In cyclic code, the syndrome vector polynomial is given as,

S (P) = remainder (y (p)/ G (P))

Y(P) is received vector polynomial and G (p) is generator polynomial.

74. What is BCH Code ?

BCH codes are most extensive and powerful error correcting cyclic code. The decoding of BCH coder is comparatively simpler. For any positive integer ‘m’ and ‘t’ , there exists a BCH code with following parameters :

Block length $n = 2^{m-1}$

No. of parity check bits : $n-k \leq mt$

Minimum distance : $d_{min} \geq 2t+1$

75. What are the conditions to satisfy the hamming code.

1) No. of Check bits $q \geq 3$
2) Block length $n = 2^q - 1$
3) No of message bits $K = n-q$
4) Minimum distance $d_{min} = 3$

76. Define code word & block length.

The encoded block of ‘n’ bits is called code word. The no. of bits ‘n’ after coding is called block length.

77. Give the parameters of RS codes .

Reed Solomon codes. These are non binary BCH codes.

Block length = n $=2^m -1$ symbols

Message size : k symbols

Parity check size : n-k= 2t symbols

Minimum distance , dmin =2t +1 symbols.

78. Why RS codes are called maximum distance separable codes ?

(n,k) Linear block code for which the minimum distance equals n – k + 1 is called maximum distance separable codes. For RS code minimum distance equals n – k + 1 so it is called as maximum distance separable codes.

79. What are Golay codes ?

Golay code is the (23, 12) cyclic code whose generating polynomial is,

$$G(p) = P^{11}+P^9+p^7+P^6+p^5+p+1$$

This code has a minimum distance of dmin=7. This code can correct upto 3 errors. It is perfect code.

80. What are the advantages of cyclic codes ?

1. Encoders and decoders for cyclic codes are simple
2. Cyclic codes also detect error burst that span many successive bits.

UNIT V
SPREAD SPECTRUM MODULATION

81. Define pseudo-noise (PN) sequence.

A pseudo-noise sequence is defined as a coded sequence of 1s and Os with certain autocorrelation properties. It is used in spread Spectrum communications. It is periodic in that a sequence of 1s and 0s repeats itself exactly with a known period.

82. What does the term catastrophic cyclic code represent ?

'000' is not a state of the shift register sequence in PN sequence generator, since this results in a catastrophic cyclic code i.e once the 000 state is entered, the shift register sequence cannot leave this state.

83. Define a random binary sequence.

A random binary sequence is a sequence in which the presence of a binary symbol 1 or 0 is equally probable.

84. State the balance property of random binary sequence.

In each period of a maximum length sequence, the number of 1s is always one more than the number of 0s. This property is called the balance property.

85. Mention about the run property.

Among the runs of 1s and 0s in each period of a maximum length sequence, one half the runs of each kind are of length one, one fourth are of length two, one eighth are of length three, and so or as long as these function represent meaningful numbers of runs. This property is called the run property.

86. Give the correlation property of random binary sequence.

The autocorrelation function of a maximum length sequence is periodic and binary valued. This property is called the correlation property.

87. Mention the significance of spread spectrum modulation.

An important attribute of spread-spectrum modulation is that it can provide protection against externally generated interfering (jamming) signals with finite power. The jamming signal may consist of a fairly powerful broadband noise or multitone waveform that is directed at the receiver for the purpose of disrupting communications. Protection against jamming waveforms is provided by purposely making the information bearing signal occupy a bandwidth far in excess of minimum bandwidth necessary to transmit it.

88. What is called processing gain ?

Processing Gain (PG) is defined as the ratio of the bandwidth of spread message signal to the bandwidth of unspreaded data signal ie).

$$\text{Processing Gain} = \frac{\text{BW (spreaded signal)}}{\text{BW (Unspreaded signal)}} =$$

89. What is called jamming effect ?

In the frequency band of the interest, somebody else transmits the signals intentionally since these signals the in the frequency band of transmission, they interface the required signal. Hence it becomes difficult to detect the required signals. This is called jamming effect.

90. What is Anti jamming ?

With the help of spread spectrum method, the transmitted signals are spread over the mid frequency band. Hence these signals appear as noise. Then it becomes difficult for the jammers to send jamming signals. This is called antijamming.

91. What are the three codes used for the anti jamming application ?

1. Golay code (24, 12)
2. Expurgated Golay (24, 11)
3. Maximum length shift register code.

92. What is called frequency hop spread spectrum ?

In frequency hop spread spectrum, the frequency of the carrier hops randomly from one frequency to another frequency.

93. What is slow frequency hopping?

If the symbol rate of MFSK is an integer multiple of hop rate (multiple symbols per hop) then it is called slow frequency hopping

94. What is fast frequency hopping?

If the hop rate is an integer multiple of symbol rate (multiple hops per symbol) then it is called fast frequency hopping.

95. What are the two function of fast frequency hopping?

1. Spread Jammer over the entire measure of the spectrum of Txed signal.
2. Retuning the Jamming signal over the frequency band of Txed signal.

96. What are the features of code Division multiple Accesses?

1. It does not require external synchronization networks.
2. CDMA offers gradual degradation in performance when the no. of users is increased But it is easy to add new user to the system.
3. If offers an external interference rejection capability.

97. What is called multipath Interference?

The interference caused by the interfacing of the signal form the indirect path with the signal of direct path is called multipath interference.

98. What is the advantage of a spread spectrum technique?

The main advantage of spread spectrum technique is its ability to reject interference whether it be the unintentional interference of another user simultaneously attempting to transmit through the channel (or) the intentional interference of a hostile transmitter to jam the transmission.

99. What is called frequency hop spread spectrum ?

In frequency hop spread spectrum, the frequency of the carrier hops randomly from one frequency to another frequency.

100 .What is slow frequency hopping?

If the symbol rate of MFSK is an integer multiple of hop rate (multiple symbols per hop) then it is called slow frequency hopping

Essay Questions (Part B)

1. Derive the power spectral Density of a Synchronous Data pulse stream generated by a Binary, Zero mean, Wide Sense Stationary Sequence.

    i. Define mean. Auto correlation and wide sense stationary
    ii. Derive the expression of (Sa) f
    iii. Derive the expression of (Sp) f
    iv. Combine the values of (Sa) f and (Sp) f
    v. Make the necessary approximations.

2. Derive the power spectral Density of a Synchronous Data pulse stream generated by a Binary, Zero mean, Cyclostationary Sequence.

    i. Define mean. Auto correlation and Cyclostationary
    ii. Derive the expression of (Sa) f
    iii. Derive the expression of (Sp) f
    iv. Combine the values of (Sa) f and (Sp) f
    v. Make the necessary approximations.

3. Derive the power spectral Density of a Generalized M-ary Markov Source
   i. Define mean. Auto correlation and Markov source
   ii. Derive the expression of (Sa) f
   iii. Derive the expression of (Sp) f
   iv. Combine the values of (Sa) f and (Sp) f
   v. Make the necessary approximations.

4. Explain in detail about scalar communication and obtain the probability of M-ary Scalar Receiver starting from the implementation of M-ary Scalar Receiver.
   i. Draw the Block Diagram of M-ary Scalar Receiver.
   ii. Do the implementation of M-ary Scalar Receiver.
   iii. Find the probability of Error of M-ary Scalar Receiver.
   iv. Draw the signal space diagram

5. Explain in detail about vector communication and obtain the probability of Binary Vector Receiver starting from the implementation of Binary Vector Receiver.
   i. Draw the Block Diagram of Binary Vector Receiver.
   ii. Do the implementation of Binary Vector Receiver.
   iii. Find the probability of Error of Binary Vector Receiver.
   iv. Draw the signal space diagram

6. Explain the construction of Block Code and explain how error syndrome is calculated
   i. Representation of Block Code.
   ii. Generator Matrix.
   iii. Generation of Codewords.
   iv. Generation of Parity Check Matrix.
   v. Calculation OF Error Syndrome.

7. Explain in detail about Orthogonal Codes, Biorthogonal Codes and Transorthogonal Codes
   i. Definitions of codes,
   ii. Formation of orthogonal codes.
   iii. Formation of Biorthogonal Codes.
   iv. Formation of Transorthogonal Codes.

8. Explain in detail about Shannon Coding Theorem.
   i. Statement of Theorem.
   ii. Discussion of theory.
   iii. Derivation of the Theorem.

9. What is Spread Spectrum Techniques Explain in detail about Direct Sequence Spread Spectrum Techniques with necessary diagrams?
   i. Concept of Spread Spectrum Techniques

ii. Block Diagram Representation.
iii. Waveform at all stages of the system.
iv. Derivation of processing Gain.

10. What is Frequency Hopping? Explain the different types of frequency hopping with necessary diagrams.
    i. Concept of frequency hopping.
    ii. Explanation of slow frequency hopping
    iii. Explanation of Fast frequency hopping
    iv. Block Diagrams and waveform

11. Explain in detail about Golay Codes, Reed Solomon Codes and BCH Codes.
    i. Definition of Golay Codes, Reed Solomon Codes and BCH Codes.
    ii. Explanation of Golay Codes
    iii Explanation of Reed Solomon Codes
    iv Explanation of BCH Codes.

12. Explain in detail about Binary Phase Shift Keying and obtain an expression for its probability of error.
    i. Block Diagram of Transmitter and Receiver.
    ii. Explanation of Transmitter and receiver.
    iii. Signal Space Diagram
    iv. Calculation of Probability of Error

13. Explain in detail about Quadrature Phase Shift Keying and obtain an expression for its probability of error.
    i. Block Diagram of Transmitter and Receiver.
    ii. Explanation of Transmitter and receiver.
    iii. Signal Space Diagram
    iv. Calculation of Probability of Error

14. Explain in detail about Minimum Shift Keying and obtain an expression for its probability of error.
    i. Block Diagram of Transmitter and Receiver.
    ii. Explanation of Transmitter and receiver.
    iii. Signal Space Diagram0
    iv. Calculation of Probability of Error

15. Explain in detail about the optimum demodulation of Digital signals in the presence if ISI and AWGN. Also, explain about the various equalization techniques.
    i. Concept of ISI and AWGN
    ii. Derivation of ISI.
    iii. Ideal Solution and Practical Solution.
    iv. Types of Equalization.

16. Explain in detail about the operation of Non Coherent Receivers in the presence of Random Phase Channel and implement the receiver.
    i. Concept of Non Coherent Receivers.
    ii. Derivation of probability of Error.
    iii. Implementation of the Receiver.
    iv. Waveforms

17. Explain in detail about the operation of Non Coherent Receivers in the presence of Random amplitude and phase Channel and implement the receiver.
    a. Concept of Non Coherent Receivers.
    b. Derivation of probability of Error.
    c. Implementation of the Receiver.
    d. Waveforms

18. Explain in detail about the operation of Optimum Receivers in Rayleigh Channel and implement the receiver.
    a. Concept of Raleigh Receivers.
    b. Derivation of probability of Error.
    c. Implementation of the Receiver.
    d. Waveforms

19. Explain in detail about the operation of Optimum Receivers in Rician Channel and implement the receiver.
    a. Concept of Rician Receivers.
    b. Derivation of probability of Error.
    c. Implementation of the Receiver.
    d. Waveforms

20. Explain in detail about the operation of Partially Coherent Receivers in the presence of Random Phase Channel and implement the receiver.
    a. Concept of Partially Coherent Receivers.
    b. Derivation of probability of Error.
    c. Implementation of the Receiver.
    d. Waveforms

21. a. Explain in detail about In phase and Quadrature Modulation systems.
    b. With necessary diagrams explain the operation of Quadrature Amplitude Modulation systems.
    a. In phase and Quadrature Modulation systems.
        - Block Diagram
        - Derivation
        - Explanation
    b. Quadrature Amplitude Modulation systems.

- Draw the Block Diagram
- Derivation
- Explanation

22. Draw the code tree of a Convolutional code of code rate r=1/2 and Constraint length of K=3 starting from the state table and state diagram for an encoder which is commonly used.
    a. Draw the state Diagram.
    b. Draw the state Table.
    c. Draw the code Tree

23. Draw the trellis diagram of a Convolutional code of code rate r=1/2 and Constraint length of K=3 starting from the state table and state diagram for an encoder which is commonly used.
    a. Draw the state Diagram.
    b. Draw the state Table.
    c. Draw the trellis diagram

24. Decode the given sequence 11 01 01 10 01 of a convolutional code with a code rate of r=1/2 and constraint length K=3, using viterbi decoding algorithm.
    a. Draw the state Diagram.
    b. Draw the state Table.
    c. Draw the code Tree
    d. Decode the given sequence using trellis diagram

25. Explain in detail about Continuous Phase Frequency Shift Keying and obtain an expression for its probability of error.
    a. Block Diagram of Transmitter and Receiver.
    b. Explanation of Transmitter and receiver.
    c. Signal Space Diagram
    d. Calculation of Probability of Error

26. Derive the power spectral Density of a Synchronous Data pulse stream generated by a Binary, non-zero mean, Cyclostationary Sequence.

    a. Define mean. Auto correlation and wide sense stationary
    b. Derive the expression of (Sa) f
    c. Derive the expression of (Sp) f
    d. Combine the values of (Sa) f and (Sp) f
    e. Make the necessary approximations.